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APPLICATION FOR LETTERS PATENT

TITLE: SOUND GENERATING DEVICE AND METHOD FOR
A MOBILE TERMINAL OF A WIRELESS
TELECOMMUNICATION SYSTEM

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**Sound generating device and method for a mobile terminal of a wireless
telecommunication system**

10 The present invention relates to a sound generating device and a sound generating method for a mobile terminal of a wireless telecommunication system.

15 Mobile terminals of wireless telecommunication systems, as e.g. the GSM or the UMTS system, usually comprise a sound unit for generating and outputting sounds, as e.g. melodies, to indicate an incoming phone call, a received message, a preset time and date, or the like to a user. In the first mobile terminals on the market, the sound generating unit only comprised a small number of prestored melodies and/or tones, from which the user could select his or her preferred ringer or alarm signal. Some of the more recent mobile terminals of wireless telecommunication systems present the possibility of creating an own melody within the limited range of one or two octaves, whereby the tone for the melody composed by the user is preset and cannot be selected by the user.

20 The object of the present invention is therefore to provide a sound generating device and a sound generating method for a mobile terminal of a wireless telecommunication system, which enable a user to choose sounds and their pitch and thus to compose melodies in a simple but flexible way.

25 The above object is achieved by a sound generating device for a mobile terminal of a wireless telecommunication system according to claim 1, comprising memory means for storing waveforms, each waveform corresponding to a sound and each waveform comprising a predetermined number of samples, selecting means for selecting a sound and a pitch for said sound to be generated, calculating means for calculating on the basis of a preset calculation rule, a sound table from the samples of the waveform of a selected sound, reading means for reading out a part of the samples from said calculated sound table depending on the selected pitch for the sound, and output means for outputting a sound on the basis of the part of samples read out from said reading means.

30 The above object is further achieved by a sound generating method for a mobile terminal of a wireless telecommunication system according to claim 11, comprising the steps of selecting a sound and a pitch for a sound to be generated from stored

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5 waveforms, each stored waveform corresponding to a sound and each waveform comprising a predetermined number of samples, calculating, on the basis of a preset calculation rule, a sound table from the samples of the waveform of a selected sound, reading out a part of the samples from the calculated sound table depending on the selected pitch for said sound, and outputting a sound on the basis of the read out part of samples.

10 The sound generating device and the sound generating method according to the present invention enable a user to choose a sound to be generated and the pitch, in which the sound should be outputted, in a simple and flexible way. Thus, different melodies can be created and composed without necessitating large processing power. Further, the sound generating device and the sound generating method of the present invention allow a simple and cost-effective implementation in a mobile terminal of a wireless telecommunication system. The sounds stored in form of waveforms in the memory
15 means can e.g. be the sound of a musical instrument, a human voice, an animal's sound or any other possible sound. Each sound has a certain typical frequency distribution, as e.g. a fundamental frequency and higher harmonics. Digitally sampling such a frequency distribution with a predetermined number of samples gives a waveform. Processing the stored waveforms according to the present invention thereby bases on a
20 sampling rate converter principle.

Advantageously, each waveform consists of one period of samples of a frequency distribution of the sound to be generated. Further advantageously, each waveform consists of the predetermined number of 51 samples. The number of 51 samples is an
25 optimised value in view of a minimalisation of resources for the implementation and a minimalisation of the frequency error in the outputted sound.

Further advantageously, the calculating means calculates the sound table on the basis of an interpolation calculation. This means that between respective two adjacent samples
30 of the stored waveform an interpolation calculation is performed so that the waveform is upsampled. In this case, the number of calculated interpolated samples between two adjacent samples of the stored waveform depends on the selected pitch for the sound to be generated. In this way, an optimisation between implementation resources and frequency error can be achieved. Advantageously, the number of calculated interpolated
35 samples is the same for each note of an octave, but decreases with ascending octaves. Hereby, for each note within one octave the same calculated sound table can be used so that the required processing is significantly reduced.

Further advantageously, the reading means reads out every n-th sample from the sound table, n being an integer number. This allows a further reduction of the processing required before outputting the selected sound. Advantageously, the number n depends on the selected pitch for the sound to be generated. Advantageously, the number n increases with ascending notes within an octave, but is the same for each respective note in the different octaves. This further decreases the storage and processing resources, since only a single number n has to be stored for each note in an octave. In different octaves, the same number n for each tone can be re-used. Further advantageously, the samples are read out from the sound table with a rate of about 8 kHz so that the highest frequency of the outputted sound can be 4 kHz.

The present invention is explained in more detail in the following description of a preferred embodiment thereof in relation to the enclosed drawings, in which

Figure 1 shows schematically a mobile terminal of a wireless telecommunication system comprising a sound generating device according to the present invention, and

Figure 2 shows a diagram explaining the generation of a sound according to the present invention.

Figure 1 shows schematically a mobile terminal 1 for a wireless telecommunication system, as e.g. the GSM or the UMTS system. It is to be noted that in figure 1 only elements necessary for explaining the present invention are shown, whereas other elements necessary for the normal operation of the mobile terminal 1 in a wireless telecommunication system, such as antenna, HF part, modulator, demodulator, coder, decoder, etc. are not shown for the sake of clarity.

The mobile terminal 1 comprises a loudspeaker 2 for outputting audible signals to indicate an incoming call, a received message, a preset time and date or the like to a user. The loudspeaker 2 can either be the same loudspeaker which is used to output speech signals during a conversation to the user, or can be a separated loudspeaker only for outputting alarm or indication signals. In the second case, the loudspeaker 2 can be implemented as a simple and cheap element.

The mobile terminal 1 further comprises an input means 3, which can e.g. be the normal keyboard of the mobile terminal. By choosing a corresponding menu function of the mobile terminal, a user can choose or select a sound and a pitch for a sound which he uses to be generated and used as an alarm signal, indication signal or the like. The

- input means 3 is connected to a control unit 4 which controls the generation of the selected sound. The control unit 4 is connected to a memory means 5, which is e.g. a non-volatile memory, in which waveforms corresponding to sounds are stored. Each waveform corresponds to a certain sound, as e.g. the sound of an instrument, the sound of a human voice, the sound of an animal or the like, and comprises a predetermined number of samples. In the preferred embodiment, each waveform comprises a single period of a frequency distribution of a corresponding sound in form of a number of 51 digitised samples. The number of 51 samples for each waveform stored in the memory means 5 together with the specific values given below in tables 1 and 2 represent optimal values in view of an optimisation of processing resources in the mobile terminal and frequency error for the outputted sound. In this context it has to be noted that a word length of 8 bit is sufficient for the purpose of the output of alarm signals, indication signals, ringer tones and the like.
- The user intending to select a sound and a pitch for an alarm signal or the like may input the selected sound or pitch by means of the input means 3. Alternatively, a sound and a pitch for a sound to be generated may be selected through any other selecting means, e.g. via transmission of a corresponding message from a base station of the wireless terminal communication system or the like. After the selection of a sound and a pitch for the sound to be generated, e.g. via input at the input means 3, a corresponding information is transmitted to the control unit 4. The control unit 4 comprises a calculating means 6 for calculating, on the basis of a preset calculation rule, a sound table from the samples of the waveform of a selected sound. In the preferred embodiment, the preset calculation rule is an interpolation calculation, whereby the number of calculated interpolated samples between two adjacent samples of the waveform depends on the selected pitch for the sound to be generated. Thereby, the number of calculated interpolated samples is the same for each note of an octave, but decreases with ascending octaves. Optimized values for the number of interpolated values per sample and a correspondingly resulting number of samples for the sound table in each octave are given in the following table 1.

Octave	c-h 262-494 Hz	c'-h' 524-988 Hz	c''-h'' 1048-1976 Hz	c'''-h''' 2096-3952 Hz
No. of interpolated values per sample	47	23	11	5
No. of samples in sound table	2448	1224	612	306

Table 1

For the first octave c-h (262-494 Hz), the number of interpolated values per sample is 47, so that a total of $(48 \times 51 =)$ 2048 samples are comprised in each respectively calculated sound table. For the second octave c'-h' (524-988 Hz), 23 values are interpolated per sample, so that a total $(24 \times 51 =)$ 1224 samples are comprised in each calculated sound table. For the third octave c''-h'' (1048-1976 Hz), 11 values are interpolated between two adjacent samples, so that a total of (12×51) 612 samples are comprised in each calculated sound table. For the third octave c'''-h''' (2096-3952 Hz), 5 values are interpolated between two adjacent samples, so that a total of $(6 \times 51 =)$ 306 samples are comprised in each calculated sound table. The number of interpolated values between two adjacent samples for each octave is about half of the value of the number of interpolated samples between two adjacent samples for the preceding octave. It is to be noted that higher pitches require a lower number of samples for a sound table due to the 4 kHz limitation (end of the fourth octave), so that memory space and processing power can be reduced in the higher frequency range. The 4 kHz limitation and the consequent sampling frequency of 8 kHz are the frequency limit and the sampling frequency, respectively, of digital mobile phones. Choosing the same frequency limit and sampling frequency has the advantage that the D/A converter of the mobile terminal intended for the output of the voice signal can also be used for the melody generation according to the present invention.

As the above-mentioned interpolation calculation for the calculation of the sound table, a linear, a polynomic or any other interpolation calculation may be used. Instead of an interpolation calculation, another way of calculating intermediate values between two adjacent samples of a selected waveform may be used.

A sound table calculated by the calculation means 6 is stored in a volatile memory 7 of the control unit 4, as e.g. a RAM memory. Thereafter, a sound table stored in the volatile memory 7 is read out by a reading means 8 also comprised in the control unit 4. Hereby, the reading means 8 reads out only a part of the samples of a sound table stored in the volatile memory 7. Thereby, the number of samples read out by the reading means 8 depends on the selected pitch for the sound to be outputted. In the preferred embodiment, the reading means 8 reads out every n-th sample from a sound table stored in the volatile memory 7, n being an integer number. Particularly, the number n increases with ascending notes within an octave, but is the same for each respective note in the different octaves, as shown in the following table 2.

note	c	c#	d	d#	e	f	f#	g	g#	a	a#	h
n	80	85	90	95	101	107	113	120	127	135	143	151

Table 2

5 For each note c in each octave, n equals 80, for each note c# in each octave, n equals 85 and so forth. The reading means 8 reads out every n-th sample from a sound table stored in the volatile memory 7 preferably by using a register as index to the sound table and incrementing the register by n for every read out value with subsequent reduction of the result modulo the length of the sound table so that periodic output

10 sounds can be generated from a single period sound table. The reading means 8 uses a readout rate of about 8 kHz for reading out every n-th sample from a sound table stored in the volatile memory 7 and supplies the read samples to an output buffer 9 of the control unit 4. From the output buffer 19, the samples are supplied to an envelope means 10 for subjecting the samples to an amplitude envelope function in order to

15 obtain a natural sound. The amplitude envelope function comprises e.g. an attack, a decay, a sustain and a release parameter upon which the natural sound of the generated waveform is based. These parameters can either be prestored or can be set by a user or by any other measure. The envelope means 10 is also part of the control unit 4. The correspondingly processed digital samples output from the envelope means 10 are

20 supplied to a digital-to-analog converter 11 which converts the digital samples into analog signals. From the digital-to-analog converter 10, the analog signals are supplied to a power amplifier 12, which supplies the amplified signals to the loudspeaker 2, which in turn outputs the amplified signals as the selected sound in the selected pitch.

25 If more than one voice or sound is to be generated and output at a time, the above described processing is carried out in parallel in as many instances as voices are to be obtained. For example, a sound consisting of different voices and/or instruments can thereby be generated. In such a case, the obtained digital values are added and then output to the digital-to-analog converter 11 before being amplified and output.

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The implementation of the sound generation according to the present invention as schematically shown in figure 1 can be implemented in different ways. One possibility is the implementation in special hardware like an integrated circuit containing a clock generator, registers for the waveform tables, logic circuitry for the interpolation and

35 reading of the tables and digital-to-analog converters. Another possibility is to implement the sound generation functions in the digital signal processor of a mobile

phone and use the digital-to-analog converter of the mobile phone used for outputting the normal voice signal to output the generated waveform or generated waveforms.

5 The numbers given in table 1 and table 2 and the number of 51 samples per stored waveform are optimised value in view of the reduction of memory and processing power and frequency error. For the given values, the frequency error is lower than 0.27% between 262 Hz (note c) and 3952 Hz (note h'''). Further, the given values lead to optimised frequency ratios for each single octave of the audible region of a human ear. Optimised frequency ratios for each octave means here that the ratio of $\sqrt[12]{2}$ for the
10 frequencies of adjacent notes is realised very closely.

Figure 2 shows a diagram for explaining the sound generation according to the present invention by means of the example of the note d of the first octave, i.e. a note with the frequency 294 Hz, e.g. of the instrument Trombone. It is assumed that a single period
15 of a waveform of the sound of the instrument Trombone is stored in the memory means 5 of the mobile terminal 1 shown in figure 1. The waveform comprises 51 samples, which are read out and processed by the calculating means 6 of the control unit 4 by calculating 47 interpolated values between each two adjacent samples S1, S2, ..., S51, so that a sound table with a total length of 2448 samples 1, 2, 3, ..., 2448 is created.
20 The sound table is then stored in the volatile memory 7. The reading means 8 reads out every 90th sample from the sound table stored in the volatile memory 7 with a repetition rate of 8 kHz. As for example shown in figure 2, the samples 1, 91, 181, 271, ..., 2431, 73, 163, 253, ..., 2413, 55, 145, and so on are read out in a circular way so that a periodic Trombone tone with the note d is generated. If another sound
25 like e.g. a flute or a violin is desired, the same procedure described above is performed with another set of 51 samples belonging to the chosen sound. Thus, the present invention allows to generate periodic signals with the frequencies of musical tones from stored single periods of waveforms in a simple but effective way.